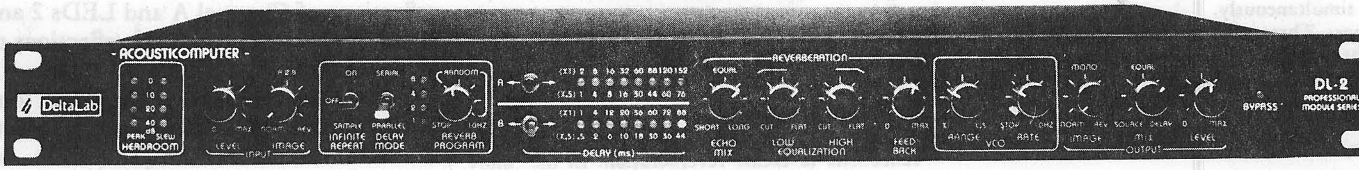
Delta Lab DL2 Acousticomputer



by Dave Hastilow

The small studio operator or musician is hit by two facts of life. The first one is that producers often take their bands to well-equipped studios, and the second is that equipping a studio or PA rack costs a lot of money. Especially a rack full of what are fast becoming the 'obligatory ancillaries' such as digital delay, flanger, etc etc.

However, well-equipped studios have to update their hourly rates to cover costs so not-so-well-off operators can score a double direct hit if they find one piece of equipment which will do the work of a rack full and is moderately enough priced to justify keeping the hourly rate down. The other fact of life is that producers always have their eyes open for less expensive well-equipped studios, so you might score another hit if a producer looks in your direction after he's heard a few of your demos or happens to be at your gig.

Delta Lab, who have the interests of the small as well as the big wigs at heart, have had their thinking caps on and have produced what appears to be one of the best ideas in a long time – the *DL2 Acousticomputer*. Basically what it all boils down to is that they've taken a 19in × 1¾in × 8in box and filled it with enough electronics to give the user access to every type of effect from digital delay, ADT, flanging, reverberation, ambient reverberation, rear channel ambience extraction, digital sample flanging, digital random flanging, digital random sample flanging, echo, cardboard tube echo, slapback, rotating speaker, vibrato, chorus vibrato, simultaneous reverb phasing, simultaneous echo phasing and all of the effects in between the above knob settings (which can be pretty mind boggling), and they're all in stereo. How does it work? Read on – you might be able to afford it.

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Basically, and in the simplest words available at this time on Sunday evening, the two input channels A and B enter the device and are each routed through two independent side chains which may remain independent or may interact with each other depending upon which mode of operation is selected. Channel A may be delayed from .5 to 152 ms and Channel B from .25 to 88 ms with the *DL-2* in the parallel mode. However, in the 'serial' mode the inputs are mixed and processed first through Channel A and then Channel B. The output of Channel B is delayed by the sum of Channel A and Channel B indicators. Also the Channel B data is twice processed. I told you it was simple.

In practical terms imagine the situation where you are trying to get an ambient guitar sound from a dry miked guitar track. Had the guitar amp been in the middle of the empty studio or room and had the mic been stereo the sound waves hitting the mic would have consisted of the direct sound from the amp mixed with multiphase delayed sound waves arriving a short time later after being reflected off the walls, floor and ceiling. This is essentially what the *DL-2* is doing in its most basic mode. Channel A delay and Channel B delay are in effect the walls around the amp, the reflections from the walls are mixing together somewhere in the middle of the room, and is the case with the *DL-2* when the Image control is in the A/B position and the resultant blend of dry plus delayed information gives the ambient effect. The reason Delta Lab have labelled the *DL-2* the *Acousticomputer* is that every delay tap within the device has a binary address code or number between 0 and 15, and the *DL-2* can select any number and regenerate it or it can scan the whole series of fifteen addresses and select randomly at a given sample rate determined by the position of the reverb knob. This is a brief description of what the *DL-2 Acousticomputer* does in its simplest mode and it is possibly the closest electronic approach to artificial ambience creation short of playing the guitar track back through a speaker in the studio and recording it.

To complete the picture the *Acousticomputer* can not only regenerate random selected short and long delayed signals but also feed back or sustain them to simulate the build-up of what is described in books on acoustics as 'equilibrium intensity', where the direct and indirect sounds from a sound source have built up to their highest level. Sound waves bounce around a room until the energy within them has been dissipated. While they've been bouncing they will have had their frequency content altered considerably by the acoustic properties of the materials lining the walls of the studio. This effect is simulated in the *Acousticomputer* by adjusting the equalisation of the signals which are being fed back to the input for regeneration. The equalisation controls consist of high and low pass filters.

The front panel can be divided into six sections which are further sectionalised by the nature of the controls under these headings. Now that I have described what the *Acousticomputer* does I hope that these individual controls will make more sense.

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INPUT SECTION

Headroom indicator: Two columns of LEDs indicating peak and slew level between −40db and 0db.

Level Control: Adjusts input level of Channel A and B simultaneously.

Image: Three settings – *Norm:* Signals pass straight through to their respective channels. *A Ø B:* Each channel is mixed with a multiphase signal from the other channel to extract ambience. *Rev:* Signals are directed to opposite channel (Stereo reverse).

PROGRAM

Sustain: In essence an infinite repeat control. When *on* the input is diconnected and data is recirculated indefinitely with no degration. In the *sample* mode the on/off function switches at a rate set by the VCO. New data is loaded into the register in the 'off' function and circulated in the 'on' function. This facility makes possible a new effect – 'Digital Step Flanging'. The sustain function may be switched on by a foot pedal.

Mode: Serial – Inputs are mixed through Channel A and then B. The output of Channel A is delayed by the Channel A indication. The output of Channel B is delayed by the sum of the channel A and B indications. Also the Channel B indications are twice processed.

Parallel – Each channel is mixed via the input image control and processed independently. The reverberation will be interrelated.

Reverb: As already mentioned, there are sixteen reverberation programs stored in the *DL 2* which create effects varying between sensations of spaciousness, flutter, and tubey effects depending upon the settings of the Time Base Generator, Reverberation, and Mix controls. The programs are selected by the reverb control.

Manual – Allows user to select delay time settings in each channel.

Hold – Locks in program displayed

Select – Program changes in a slow counting manner until the knob is rotated to the hold position

Random – Program changes at a rate determined by position of knob

DELAY

Moving the switches forward or backward respectively increase or decrease the initial delays in each channel by the amount indicated in figures above and below the Light Emitting Diodes. Short delays are used for flanging and long delays for spatial or echo effects. In the parallel mode channel B behaves in the same manner as channel A but in the serial mode the initial delay of channel B output will be the sum of the delays indicated by both the Channel A LED and Channel B LED. This means that Channel B becomes Channel A + B.

REVERBERATION

The reverb mix control mixes the long and short reflections to create reverberation. In the 'short' position only short reflections will be regenerated. In the 'long' position only long-delayed reflections are regenerated. In the 'equal' position equal amounts of short and long are fed back. In practical terms the 'short' position corresponds to the reverberation of a small room and so on.

Equalisation: Low control rolls off low frequencies in the regenerated delayed signals. At full cut passes frequencies above 800 Hertz only. High control rolls off everything above 1.5kHz.

Feedback: Determines the amplitude of signals being fed back. At max +, signals are fed back in phase with input. At zero, output has no reverberation. At max −, reflections are fed back out of phase with input.

TIME BASE GENERATOR

Delay Factor: Varies the basic clock frequency which determines the delay; thus acting as a variable delay multiplier. Variable between ×1 and ×4.

VCO Rate and Depth: Rate variable between 0 and 10 Hz. Depth control varies amplitude of VCO frequency thus making deep flange and pitch bending possible.

OUTPUT

Image: As with the input image control allowing subtle stereo effects to be created.

Mix: Allows the source (bypass) or the delayed signals only to appear at the outputs or a mix of both when the knob is set in the equal position.

The *DL 2* can be bypassed by using a footpedal which plugs into the back of the unit.

In the Program section of the front panel are set four LEDs which are numbered 1 2 4 8 upwards. These indicate the binary address codes for the Channel A and Channel B short and long initial reflections which are recirculated to create reverberation. LEDs 8 and 4 indicate the short and long initial reflections of Channel A and LEDs 2 and 1 the short and long initial reflections of Channel B.

The precise amount of short and long initial reflections recirculated to create the final reverberation pattern is determined by the position of the Reverb Mix control.

In operation the *Acousticomputer* does everything it says it can and more. It is not possible to describe in words the effects created when the knobs are in the intermediate settings and it is advisable to make slight rather than large adjustments to the knobs when looking for effects to prevent overloading other equipment which may be interfaced with it. It is possible to set up quite a simple program and flick the parallel/series and sustain switches only to be knocked out by something else, so beware of Acousticomputeritis, or sidetracking! The chorus effect is quite astounding; so is the rotating speaker, and the reverberation is similar in texture to that provided by a reasonably-priced spring. The input and output connections are balanced and unbalanced which makes for good mobility between studio and stage and three sockets allow for the sustain and bypass modes to be switched by footpedal or for insertion of an external control voltage of between 0 and 10v. Provision is also made for an optional memory module, which increases the *DL 2* memory capability to 2 seconds, to be inserted via a 5 pin Switchcraft connector. The increased delay time will not degrade any of the *DL 2 Acousticomputer* functions.■